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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/532,593	08/18/2005	Stuart Charles Wray	038665.56183US	4830

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EXAMINER	
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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/532,593	WRAY ET AL.	
	Examiner	Art Unit	
	Christopher Crutchfield	2466	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE _3_ MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on _22 January 2010_.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) _1,2,7-10 and 12-14_ is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) _1, 2, 7-10 and 12-14_ is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Continued Examination Under 37 CFR 1.114

1. A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on 6 May 2010 has been entered.

Claim Rejections - 35 USC § 103

2. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

3. The factual inquiries set forth in *Graham* v. *John Deere Co.*, 383 U.S. 1, 148 USPQ 459 (1966), that are applied for establishing a background for determining obviousness under 35 U.S.C. 103(a) are summarized as follows:

1. Determining the scope and contents of the prior art.
2. Ascertaining the differences between the prior art and the claims at issue.
3. Resolving the level of ordinary skill in the pertinent art.
4. Considering objective evidence present in the application indicating obviousness or nonobviousness.

4. This application currently names joint inventors. In considering patentability of the claims under 35 U.S.C. 103(a), the examiner presumes that the subject matter of the various claims was commonly owned at the time any inventions covered therein were made absent any evidence to the contrary. Applicant is advised of the obligation under 37 CFR 1.56 to point out the inventor and invention dates of each claim that was not commonly owned at the time a later invention was made in order for the examiner to consider the applicability of 35 U.S.C. 103(c) and potential 35 U.S.C. 102(e), (f) or (g) prior art under 35 U.S.C. 103(a).

5. **Claim 1** is rejected under 35 U.S.C. 103(a) as being unpatentable over *Qiu*, et al. (J. Qiu, H. Shao, W. Zhu and Y. Zhang, An End-to-End Probing Based Application Control Scheme for Multimedia Applications, IEEE International Conference on Multimedia, 2001, Pages 872-875) in view of *Komatsu*, et al. (US Patent No. 6, 914,900 B1).

Regarding claim 1, *Qiu* discloses a method of admission control for a continuous stream of data in packet switched networks including at least two networks that communicate with one another across a connecting network, the method comprising:

a. Determining a packet loss rate of previous transmissions to a network (Pages 872-874, Particularly Sections 2-4). (The system of *Qiu* discloses a call admission control system that performs measurement based admission control [Page 872, Section 2]. The system operates by sending continuous probes to a second network endpoint in order to determine network delay and loss characteristics [Page 873, Section 2, Left Column, First Full Paragraph]. The loss and delay characteristics are then used to determine the delay shape distributions by tracking the probability of various packet delays for various

packet loss rates [Page 873, Section 3]. For example, when the desired loss rate [i.e. state "0"] is set to 1%, the system tracks the "delay shape" [i.e. the average delay- the minimum delay] for all measurements with a loss rate less then the “0” state [i.e. all measurements with a loss rate <1%] and for all measurements with a loss state of “1” [i.e. all measurements where the loss rate exceeded 1%] [Page 873, Section 3, Particularly Fig. 2 and the Preceding Paragraph]. The intersection of the delay shape functions for the "0" and “1” state is then used to determine the low and high delay shape distribution profiles [Page 873, Section 3, Particularly Fig. 3][See also Page 873, Page 4, Particularly Equations (2) and (3) - Showing the calculation of the low and high delay shape profiles]. Whenever the system detects an incoming connection control request, it compares the delay state from the last set of network probe packets with the delay state profiles [which are based on past packet los rate and delay - i.e. the “packet loss rate of previous network transmissions”] [Pages 872-873, Section 2, Particularly Fig. 1]. If the delay shape is below the high delay shape profile but above the loss delay state profile, then the system will send a test probe to the network to determine the current network loss characteristics [i.e. the “current packet loss rate”] and will admit or reject the transmission based on if the current loss rate is above or below and admission threshold [i.e. “decide to drop an attempt based on the current packet loss rate”] [Fig. 1, and Page 872, Last Paragraph, Carried on to Page 873][See also Page 873, Last Paragraph, Carried onto Page 874]. If the delay shape is above the delay shape high profile, then the probing step is not performed, and the admission request is denied [Page 873, Fig. 1]. If the delay shape profile is below the delay shape low profile after taking into account the new flow, the system automatically admits the new flow without probing the network [Page 873, Fig. 1].)

b. Determining a current packet loss rate based on said packet loss rate of transmissions (Pages 872-874, Particularly Sections 2-4) (See (a), Supra).

c. Deciding to drop an attempt based on the current packet loss rate (Pages 872-874, Particularly Sections 2-4) (See (a), Supra)

Qiu fails to disclose the method is for call admission control and tracks packet loss rates between local area networks based on previous call loss data such that the method comprises a method of call admission control for a continuous stream of data in packet switched networks including at least two local area networks that communicate with one another across a connecting network, the method comprising determining a packet loss rate of previous calls to a local area network, determining a current packet loss rate based on said packet loss rate of previous calls and deciding to drop a call attempt based on the current packet loss rate. (i.e. The system of *Qiu* discloses that the system obtains its delay and loss data by sending periodic probes to network endpoints in other networks, as opposed to gathering delay and loss data by observing the delay and loss statistics of flows to the remote local area networks. *Qiu* also discloses a generic admission control technique, as opposed to a call admission control technique, and although a references it cites acknowledges that admission control is applied to call admission control [See for Example Page 875, Reference [6], titled “Measurement-based call admission control…”] it fails to disclose that the technique is applied directly to call admission control). In the same field of endeavor, *Komatsu* discloses the method is for call admission control and tracks packet loss rates between local area networks such that the method comprises a method of call admission control for a continuous stream of data in packet

switched networks including at least two local area networks that communicate with one another across a connecting network, the method comprising determining a packet loss rate of previous calls to a local area network, determining a current packet loss rate based on said packet loss rate of previous calls and deciding to drop a call attempt based on the current packet loss rate (Column 5, Lines 5-20, Column 7, Lines 15-27, Column 3, Line 24 and Column 8, Lines 23-30). (The system of Komatsu discloses a system that obtains delay and packet loss data by tracking the delay and packet loss statistics for past calls to a network of interest [Column 5, Lines 5-20, Column 6, Line 65 to Column 7, Line 6 and Column 7, Lines 15-27]. When a call is made, the system references the loss statistics for the last call to that IP endpoint at the same time and date from the current endpoint and if the loss rate is acceptable, the call is admitted. If the loss rate is unacceptable, the user is notified and may drop the call [Column 3, Line 24]. The call is made between two different LANs, as indicated by the fact that traffic form the IP packetizing unit passes through a router to reach its destination [Column 8, Lines 23-30].)

Therefore, since *Komatsu* discloses a cal admission control scheme for tracking delay and packet loss between two LANs by monitoring the delay and loss statistics of calls between the networks, it would have been obvious to a person of ordinary skill in the art at the time of the invention to combine the call based network tracking and admission control of *Komatsu* with the system of *Qiu* by tracking the delay and loss characteristics for a remote local area network using call statistics to that network as opposed to active probing and using the statistics to perform admission control, as taught by *Komatsu*. The motive to combine is to decrease network traffic by removing the need to perform active probing to determine the network delay profiles.

6. **Claim 2 is** rejected under 35 U.S.C. 103(a) as being unpatentable over *Qiu*, et al. (J. Qiu, H. Shao, W. Zhu and Y. Zhang, An End-to-End Probing Based Application Control Scheme for Multimedia Applications, IEEE International Conference on Multimedia, 2001, Pages 872-875) in view of *Komatsu*, et al. (US Patent No. 6, 914,900 B1), *Odom* (Odom, Cisco VOIP Call Admission Control, August 2001, Pages 1-26) and *Oran*, et al. (US Pre Grant Publication No. 2006/0034188).

Regarding claim 2, *Qiu* discloses a method of admission control for a continuous stream of data in packet switched networks including at least two networks that communicate with one another across a connecting network, the method comprising:

a. Determining a packet loss rate of previous transmissions from a first network to a second network (Pages 872-874, Particularly Sections 2-4). (The system of *Qiu* discloses a call admission control system that performs measurement based admission control [Page 872, Section 2]. The system operates by sending continuous probes to a second network endpoint in order to determine network delay and loss characteristics [Page 873, Section 2, Left Column, First Full Paragraph]. The loss and delay characteristics are then used to determine the delay shape distributions by tracking the probability of various packet delays for various packet loss rates [Page 873, Section 3]. For example, when the desired loss rate [i.e. state "0"] is set to 1%, the system tracks the "delay shape" [i.e. the average delay- the minimum delay] for all measurements with a loss rate less then the “0” state [i.e. all measurements with a loss rate <1%] and for all measurements with a loss state of “1” [i.e. all measurements where the loss rate exceeded 1%] [Page 873, Section 3, Particularly Fig. 2 and the Preceding Paragraph].

The intersection of the delay shape functions for the "0" and “1” state is then used to determine the low and high delay shape distribution profiles [Page 873, Section 3, Particularly Fig. 3][See also Page 873, Page 4, Particularly Equations (2) and (3) - Showing the calculation of the low and high delay shape profiles]. Whenever the system detects an incoming connection control request, it compares the delay state from the last set of network probe packets with the delay state profiles [which are based on past packet los rate and delay - i.e. the “packet loss rate of previous network transmissions”] [Pages 872-873, Section 2, Particularly Fig. 1]. If the delay shape is below the high delay shape profile but above the loss delay state profile, then the system will send a test probe to the network to determine the current network loss characteristics [i.e. the “current packet loss rate”] and will admit or reject the transmission based on if the current loss rate is above or below and admission threshold [i.e. “decide to drop an attempt based on the current packet loss rate”] [Fig. 1, and Page 872, Last Paragraph, Carried on to Page 873][See also Page 873, Last Paragraph, Carried onto Page 874]. If the delay shape is above the delay shape high profile, then the probing step is not performed, and the admission request is denied [Page 873, Fig. 1]. If the delay shape profile is below the delay shape low profile after taking into account the new flow, the system automatically admits the new flow without probing the network [Page 873, Fig. 1].)

b. Determining a current packet loss rate for transmissions from the first local area network to the second network (Pages 872-874, Particularly Sections 2-4) (See (a), Supra).

c. Deciding to drop connection attempt based on the current packet loss rate (Pages 872-874, Particularly Sections 2-4) (See (a), Supra).

d. Said step of determining a current packet loss rate comprises transmitting a burst of trial data from a first node in the first network through the connecting network to a second node in the second network (Pages 872-874, Particularly Sections 2-4) (See (a), Supra).

Qiu fails to disclose the method is for call admission control and tracks packet loss rates between local area networks based on previous call loss data such that the method comprises a method of call admission control for a continuous stream of data in packet switched networks including at least two local area networks that communicate with one another across a connecting network, the method comprising determining a packet loss rate of previous calls from a first local area network to a second local area network determining a current packet loss rate for calls from the first local area network to the second local area network and deciding to drop call attempt based on the current packet loss rate wherein said step of determining a current packet loss rate comprises transmitting a burst of trial data from a first node in the first local area network through the connecting network to a second node in the second local area network. (i.e. The system of *Qiu* discloses that the system obtains its delay and loss data by sending periodic probes to network endpoints in other networks, as opposed to gathering delay and loss data by observing the delay and loss statistics of flows to the remote local area networks. *Qiu* also discloses a generic admission control technique, as opposed to a call admission control technique, and although a references it cites acknowledges that admission control is applied to call admission control [See for Example Page 875, Reference [6], titled

"Measurement-based call admission control…"] it fails to disclose that the technique is applied directly to call admission control). In the same field of endeavor, *Komatsu* the method is for call admission control and tracks packet loss rates between local area networks based on previous call loss data such that the method comprises a method of call admission control for a continuous stream of data in packet switched networks including at least two local area networks that communicate with one another across a connecting network, the method comprising determining a packet loss rate of previous calls from a first local area network to a second local area network determining a current packet loss rate for calls from the first local area network to the second local area network and deciding to drop call attempt based on the current packet loss rate wherein said step of determining a current packet loss rate comprises transmitting a burst of trial data from a first node in the first local area network through the connecting network to a second node in the second local area network (Column 5, Lines 5-20, Column 7, Lines 15-27, Column 3, Line 24 and Column 8, Lines 23-30). (The system of Komatsu discloses a system that obtains delay and packet loss data by tracking the delay and packet loss statistics for past calls to a network of interest [Column 5, Lines 5-20, Column 6, Line 65 to Column 7, Line 6 and Column 7, Lines 15-27]. When a call is made, the system references the loss statistics for the last call to that IP endpoint at the same time and date from the current endpoint and if the loss rate is acceptable, the call is admitted. If the loss rate is unacceptable, the user is notified and may drop the call [Column 3, Line 24]. The call is made between two different LANs, as indicated by the fact that traffic form the IP packetizing unit passes through a router to reach its destination [Column 8, Lines 23-30].)

Therefore, since *Komatsu* discloses a cal admission control scheme for tracking delay and packet loss between two LANs by monitoring the delay and loss statistics of calls between the networks, it would have been obvious to a person of ordinary skill in the art at the time of the

invention to combine the call based network tracking and admission control of *Komatsu* with the system of *Qiu* by tracking the delay and loss characteristics for a remote local area network using call statistics to that network as opposed to active probing and using the statistics to perform admission control, as taught by *Komatsu*. The motive to combine is to decrease network traffic by removing the need to perform active probing to determine the network delay profiles.

Qiu as modified by *Komatsu* fails to disclose the step of determining a current packet loss rate comprises reflecting the burst of trial data received at the second node back to the first node, and receiving the reflected burst of trial data at the first node through the connecting network, said burst of trial data comprises a plurality of packets having a size and priority that corresponds to packets that are to be sent if the call is completed and the step of determining to drop a call attempt comprises comparing the reflected burst of trial data to the transmitted burst of trial data to determine whether transmission of a continuous stream of data can be initiated from the first node in the first local area network to the second node in the second local area network. In the same field of endeavor, *Odam* discloses the step of determining a current packet loss rate comprises reflecting the burst of trial data received at the second node back to the first node, and receiving the reflected burst of trial data at the first node through the connecting network, said burst of trial data comprises a plurality of packets having a size and priority that corresponds to packets that are to be sent if the call is completed and the step of determining to drop a call attempt comprises comparing the reflected burst of trial data to the transmitted burst of trial data to determine whether transmission of a continuous stream of data can be initiated from the first node in the first local area network to the second node in the second local area network (Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom,

Figure 4] to the server gateway in the other network/LAN. It then measures the packet loss rate of reflected packets to determine the packet loss rate of calls between the two networks [Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value]. This value, along with others is used to perform client access control. The SAA protocol may also be configured to send probe packets based on the packet size of the codec to be used in the call using RTP headers to create a packet identical in size to one that would be used in a real voice conversation. The priority [i.e. IP precedence] of the packets may also be set to match the voice packets to be sent [Page 23, SAA Probe Format].)

Therefore, since *Odam* discloses using a reflected burst of trial data with packets having a priority and size corresponding to that of packets to be sent upon connection completion, and *Qiu* as modified by *Komatsu* discloses transmitting a burst between two phones on different local area networks, it would have been obvious to a person of ordinary skill at the time of the invention to replace the loss based burst probe of *Qiu* as modified by *Komatsu* with the burst probe of *Odam*. The motive to combine is to allow for more effective burst probing by transmitting a burst probe that more accurately reflects the packets to be transmitted on the admitted connection.

Qiu fails to disclose a method wherein the first node comprises a telephone in the first local area network and the second node comprises a telephone in the second network. In the same field of endeavor, *Oran* discloses a method wherein the first node comprises a telephone in the first local area network and the second node comprises a telephone in the second network (Paragraphs 0031-0038 and 0046). (The system of *Oran* discloses a system that simulates a voice call before initiation by sending a bi-directional stream of real time protocol (RTP) no-op packets between the sending and receiving VOIP telephones [Paragraphs 0031-

0038]. The packets may be the same size as the actual media packets that are to follow [Paragraph 0046].)

Therefore, since *Oran* discloses call simulation between two telephone endpoints, it would have been obvious to combine the endpoint call simulation of *Oran* with the system of *Odom* by having the telephone endpoints transmit a trial burst of data, as taught by *Oran* and reflecting the trial burst back to the sender, as taught by *Odom*. The motive to combine is to allow the endpoints to test the connection using the endpoints as opposed to the gateway, thereby reducing the load on the gateway.

7. **Claims 7, 9, and 10** are rejected under 35 U.S.C. 103(a) as being unpatentable over *Qiu*, et al. (J. Qiu, H. Shao, W. Zhu and Y. Zhang, An End-to-End Probing Based Application Control Scheme for Multimedia Applications, IEEE International Conference on Multimedia, 2001, Pages 872-875) and *Komatsu*, et al. (US Patent No. 6, 914,900 B1) as applied to claim 1 and further in view of *Odom* (Odom, Cisco VOIP Call Admission Control, August 2001, Pages 1-26).

Regarding claim 7, *Qiu* fails to disclose the step of determining said current packet loss rate comprises transmitting a burst of trial data from a first node in the first local area network through the connecting network to a second node in the second local area network, reflecting the burst of trial data received at the second node back to the first node, and receiving the reflected burst of trial data at the first node through the connecting network. In the same field of endeavor, *Odam* discloses the step of determining said current packet loss rate comprises transmitting a burst of trial data from a first node in the first local area network through the connecting network to a second node in the second local area network, reflecting the burst of

trial data received at the second node back to the first node, and receiving the reflected burst of trial data at the first node through the connecting network (Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom, Figure 4] to the server gateway in the other network/LAN. It then measures the packet loss rate of reflected packets to determine the packet loss rate of calls between the two networks [Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value]. This value, along with others is used to perform client access control. The SAA protocol may also be configured to send probe packets based on the packet size of the codec to be used in the call using RTP headers to create a packet identical in size to one that would be used in a real voice conversation. The priority [i.e. IP precedence] of the packets may also be set to match the voice packets to be sent [Page 23, SAA Probe Format].)

Therefore, since *Odam* discloses using a reflected burst of trial data and *Qiu* as modified by *Komatsu* discloses transmitting a burst between two different local area networks, it would have been obvious to a person of ordinary skill at the time of the invention to replace the loss based burst probe of *Qiu* as modified by *Komatsu* with the burst probe of *Odam*. The motive to combine is to allow for more accurate effective burst probing by allowing the calculation of round trip loss and delay based on reflected probe packets.

Regarding claim 9, *Qiu* fails to disclose said burst of trial data comprises a plurality of packets having a size and priority that correspond to packets that are to be sent if the call is completed. In the same field of endeavor, *Odom* discloses said burst of trial data comprises a plurality of packets having a size and priority that correspond to packets that are to be sent if the call is completed (Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom, Figure 4] to the server gateway in the other network/LAN. It then measures the packet

loss rate of reflected packets to determine the packet loss rate of calls between the two networks [Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value]. This value, along with others is used to perform client access control. The SAA protocol may also be configured to send probe packets based on the packet size of the codec to be used in the call using RTP headers to create a packet identical in size to one that would be used in a real voice conversation. The priority [i.e. IP precedence] of the packets may also be set to match the voice packets to be sent [Page 23, SAA Probe Format].)

Therefore, since *Odam* discloses using a reflected burst of trial data with packets having a priority and size corresponding to that of packets to be sent upon connection completion, and *Qiu* as modified by *Komatsu* discloses transmitting a burst between two phones on different local area networks, it would have been obvious to a person of ordinary skill at the time of the invention to replace the loss based burst probe of *Qiu* as modified by *Komatsu* with the burst probe of *Odam*. The motive to combine is to allow for more effective burst probing by transmitting a burst probe that more accurately reflects the packets to be transmitted on the admitted connection.

Regarding claim 10, *Qiu* fails to disclose said step of determining a packet loss rate of previous calls comprises determining the packet loss rate from a first local area network to a second local area network. In the same field of endeavor, *Komatsu* discloses said step of determining a packet loss rate of previous calls comprises determining the packet loss rate from a first local area network to a second local area network (Column 5, Lines 5-20, Column 7, Lines 15-27, Column 3, Line 24 and Column 8, Lines 23-30). (The system of Komatsu discloses a system that obtains delay and packet loss data by tracking the delay and packet loss statistics for past calls to a network of interest [Column 5, Lines 5-20, Column 6, Line 65 to Column 7, Line 6 and Column 7, Lines 15-27]. When a call is made, the system references the loss

statistics for the last call to that IP endpoint at the same time and date from the current endpoint and if the loss rate is acceptable, the call is admitted. If the loss rate is unacceptable, the user is notified and may drop the call [Column 3, Line 24]. The call is made between two different LANs, as indicated by the fact that traffic form the IP packetizing unit passes through a router to reach its destination [Column 8, Lines 23-30].)

Therefore, since *Komatsu* discloses a cal admission control scheme for tracking delay and packet loss between two LANs by monitoring the delay and loss statistics of calls between the networks, it would have been obvious to a person of ordinary skill in the art at the time of the invention to combine the call based network tracking and admission control of *Komatsu* with the system of *Qiu* by tracking the delay and loss characteristics for a remote local area network using call statistics to that network as opposed to active probing and using the statistics to perform admission control, as taught by *Komatsu*. The motive to combine is to decrease network traffic by removing the need to perform active probing to determine the network delay profiles.

8. **Claims 8 and 14** are rejected under 35 U.S.C. 103(a) as being unpatentable over *Qiu*, et al. (J. Qiu, H. Shao, W. Zhu and Y. Zhang, An End-to-End Probing Based Application Control Scheme for Multimedia Applications, IEEE International Conference on Multimedia, 2001, Pages 872-875), *Komatsu*, et al. (US Patent No. 6, 914,900 B1) and *Odom* (Odom, Cisco VOIP Call Admission Control, August 2001, Pages 1-26) as applied to claim 7 and further in view of *Oran*, et al. (US Pre Grant Publication No. 2006/0034188).

Regarding claim 8, *Qiu* fails to disclose a method wherein the first node comprises a telephone in the first local area network and the second node comprises a telephone in the second network. In the same field of endeavor, *Oran* discloses a method wherein the first node comprises a telephone in the first local area network and the second node comprises a telephone in the second network (Paragraphs 0031-0038 and 0046). (The system of *Oran* discloses a system that simulates a voice call before initiation by sending a bi-directional stream of real time protocol (RTP) no-op packets between the sending and receiving VOIP telephones [Paragraphs 0031-0038]. The packets may be the same size as the actual media packets that are to follow [Paragraph 0046].)

Therefore, since *Oran* discloses call simulation between two telephone endpoints, it would have been obvious to combine the endpoint call simulation of *Oran* with the system of *Odom* by having the telephone endpoints transmit a trial burst of data, as taught by *Oran* and reflecting the trial burst back to the sender, as taught by *Odom*. The motive to combine is to allow the endpoints to test the connection using the endpoints as opposed to the gateway, thereby reducing the load on the gateway.

Regarding claim 14, *Qiu* fails to disclose said burst of trial data comprises a plurality of packets having a size and priority that correspond to packets that are to be sent if the call is completed. In the same field of endeavor, *Odom* discloses said burst of trial data comprises a plurality of packets having a size and priority that correspond to packets that are to be sent if the call is completed (Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom, Figure 4] to the server gateway in the other network/LAN. It then measures the packet loss rate of reflected packets to determine the packet loss rate of calls between the two networks [Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value]. This

value, along with others is used to perform client access control. The SAA protocol may also be configured to send probe packets based on the packet size of the codec to be used in the call using RTP headers to create a packet identical in size to one that would be used in a real voice conversation. The priority [i.e. IP precedence] of the packets may also be set to match the voice packets to be sent [Page 23, SAA Probe Format].)

Therefore, since *Odam* discloses using a reflected burst of trial data and *Qiu* as modified by *Komatsu* discloses transmitting a burst between two different local area networks, it would have been obvious to a person of ordinary skill at the time of the invention to replace the loss based burst probe of *Qiu* as modified by *Komatsu* with the burst probe of *Odam*. The motive to combine is to allow for more accurate effective burst probing by allowing the calculation of round trip loss and delay based on reflected probe packets.

9. **Claim 12** is rejected under 35 U.S.C. 103(a) as being unpatentable over *Qiu*, et al. (J. Qiu, H. Shao, W. Zhu and Y. Zhang, An End-to-End Probing Based Application Control Scheme for Multimedia Applications, IEEE International Conference on Multimedia, 2001, Pages 872-875) in view of *Komatsu*, et al. (US Patent No. 6, 914,900 B1), *Odom* (Odom, Cisco VOIP Call Admission Control, August 2001, Pages 1-26) and *Oran*, et al. (US Pre Grant Publication No. 2006/0034188) as applied to claim 2 and further in view of *Westphal*, et al. (US Pre grant Publication No. 2003/0165122 A1).

Regarding claim 12, *Qiu* discloses a method wherein said step of determining a current packet loss rate further comprises determining if the network is in a "good" or "bad" state based on the packet loss rate of previous transmissions and using the state of the network as determined form the packet loss rate of previous transmissions to determine whether to permit

the bust of trial data. (Pages 872-874, Particularly Sections 2-4). (The system of *Qiu* discloses a call admission control system that performs measurement based admission control [Page 872, Section 2]. The system operates by sending continuous probes to a second network endpoint in order to determine network delay and loss characteristics [Page 873, Section 2, Left Column, First Full Paragraph]. The delay and loss characteristics are tracked by the network based on historic probe data and used to determine if the network is currently in a "good" or “bad" state based on the calculated delay profiles [See claim 2, Supra for further details concerning the delay profiles]. If the system of *Qiu* determines that the network state is “bad” than all sessions are blocked, and if the network state is "good" then all sessions are allowed [Page 873, Fig. 1 and The Last Paragraph of Page 872 carried on to 873]. However, if the network state is between "good" and "bad" then the system performs further testing to determine the network state by probing the network [Page 873, Fig. 1 and The Last Paragraph of Page 872 carried on to 873]. The system of *Qiu* discloses that the system uses packet loss rate and delay as the network quality metrics, but also discloses that it is known in the art to use a variety of quality metrics including packet loss, delay, jitter and network load to determine network quality [Page 872, Right Column, First and Second Full Paragraphs]. Therefore, the system of *Qiu* teaches the concept of using a network state calculated from previous probes to determine if the network is currently in a “good”, "bad" or in-between state and only probing the network with trial burst of data if the system is in the in-between state where the system requires more current information before admitting the call.)

Qiu fails to disclose the inclusion of the success rate of previous calls in the determination of network state such that the step of deterring a current packet loss rate further comprises determining a success rate of previous calls and using the success rate of previous calls to determine whether to permit the bust of trial data. In the same field of endeavor,

Westphal discloses a success rate of previous calls and using the success rate of previous calls to determine whether to permit the bust of trial data (Paragraph 0080). (The system of *Westphal* discloses that it was known in the art at the time of the invention that the packet loss rate and the call drop rate are related network quality metrics, as the "dropping" of calls because of poor quality is directly related to the packet loss rate [See Paragraphs 0047 and 0080].)

Therefore, since *Westphal* teaches the packet loss rate and call drop rate are related network quality metrics for calls, and the system of *Qiu* discloses the use of historic network quality metrics to determine if the system should probe before admitting a connection, it would have been obvious to a person of ordinary skill in the art at the time of the invention to use the call drop rate and the packet loss rate to determine if the system should allow the network to probe as the call drop rate known in the art to be closely related to the packet loss rate of the network. The motive to combine is to improve the call admission control by taking into account quality metrics specifically associated with call based networks.

In the alternative, the system of *Qiu* can be viewed as teaching a base system in which network quality is measured using the packet loss rate of previous transmissions in order to determine if the network is to be probed. *Westphal* can be viewed as teaching a known technique of using the call drop rates, which are directly related to the packet loss rates, to determine the quality of a network connection. Therefore, a person of ordinary skill in the art would have recognized that the call drop rate could be used as a network quality metric, as the use of the call drop rate to determine network quality was a part of the ordinary capabilities of a person of ordinary skill in the art at the time of the invention and would have yielded the predictable result of providing a good network quality metric. This is particularly true given the known direct correlation between the packet drop rate used in *Qiu* and the network call drop rate (See *Westphal*, Paragraph 0047).

10. **Claim 13** is rejected under 35 U.S.C. 103(a) as being unpatentable over *Qiu*, et al. (J. Qiu, H. Shao, W. Zhu and Y. Zhang, An End-to-End Probing Based Application Control Scheme for Multimedia Applications, IEEE International Conference on Multimedia, 2001, Pages 872-875) in view of *Komatsu*, et al. (US Patent No. 6, 914,900 B1), *Odom* (Odom, Cisco VOIP Call Admission Control, August 2001, Pages 1-26) and *Oran*, et al. (US Pre Grant Publication No. 2006/0034188) as applied to claim 7 and further in view of *Westphal*, et al. (US Pre grant Publication No. 2003/0165122 A1).

Regarding claim 13, *Qiu* discloses a method wherein said step of determining a current packet loss rate further comprises determining if the network is in a "good" or "bad" state based on the packet loss rate of previous transmissions and using the state of the network as determined form the packet loss rate of previous calls to determine whether to permit the bust of trial data. (Pages 872-874, Particularly Sections 2-4). (The system of *Qiu* discloses a call admission control system that performs measurement based admission control [Page 872, Section 2]. The system operates by sending continuous probes to a second network endpoint in order to determine network delay and loss characteristics [Page 873, Section 2, Left Column, First Full Paragraph]. The delay and loss characteristics are tracked by the network based on historic probe data and used to determine if the network is currently in a "good" or “bad" state based on the calculated delay profiles [See claim 2, Supra for further details concerning the delay profiles]. If the system of *Qiu* determines that the network state is “bad” than all sessions are blocked, and if the network state is "good" then all sessions are allowed [Page 873, Fig. 1 and The Last Paragraph of Page 872 carried on to 873]. However, if the network state is between "good" and "bad" then the system performs further testing to determine the network

state by probing the network [Page 873, Fig. 1 and The Last Paragraph of Page 872 carried on to 873]. The system of *Qiu* discloses that the system uses packet loss rate and delay as the network quality metrics, but also discloses that it is known in the art to use a variety of quality metrics including packet loss, delay, jitter and network load to determine network quality [Page 872, Right Column, First and Second Full Paragraphs]. Therefore, the system of *Qiu* teaches the concept of using a network state calculated from previous probes to determine if the network is currently in a “good”, "bad" or in-between state and only probing the network with trial burst of data if the system is in the in-between state where the system requires more current information before admitting the call.)

Qiu fails to disclose the inclusion of the success rate of previous calls in the determination of network state such that the step of deterring a current packet loss rate further comprises determining a success rate of previous calls and using the success rate of previous calls to determine whether to permit the bust of trial data. In the same field of endeavor, *Westphal* discloses a success rate of previous calls and using the success rate of previous calls to determine whether to permit the bust of trial data (Paragraph 0080). (The system of *Westphal* discloses that it was known in the art at the time of the invention that the packet loss rate and the call drop rate are related network quality metrics, as the “dropping” of calls because of poor quality is directly related to the packet loss rate [See Paragraphs 0047 and 0080].)

Therefore, since *Westphal* teaches the packet loss rate and call drop rate are related network quality metrics for calls, and the system of *Qiu* discloses the use of historic network quality metrics to determine if the system should probe before admitting a connection, it would have been obvious to a person of ordinary skill in the art at the time of the invention to use the call drop rate and the packet loss rate to determine if the system should allow the network to probe as the call drop rate known in the art to be closely related to the packet loss rate of the

network. The motive to combine is to improve the call admission control by taking into account quality metrics specifically associated with call based networks.

In the alternative, the system of *Qiu* can be viewed as teaching a base system in which network quality is measured using the packet loss rate of previous transmissions in order to determine if the network is to be probed. *Westphal* can be viewed as teaching a known technique of using the call drop rates, which are directly related to the packet loss rates, to determine the quality of a network connection. Therefore, a person of ordinary skill in the art would have recognized that the call drop rate could be used as a network quality metric, as the use of the call drop rate to determine network quality was a part of the ordinary capabilities of a person of ordinary skill in the art at the time of the invention and would have yielded the predictable result of providing a good network quality metric. This is particularly true given the known direct correlation between the packet drop rate used in *Qiu* and the network call drop rate (See *Westphal*, Paragraph 0047).

Response to Arguments

11. Applicant's arguments with respect to claims 1, 2, 7-10 and 12-14 have been considered but are moot in view of the new ground(s) of rejection.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Christopher Crutchfield whose telephone number is (571) 270-3989. The examiner can normally be reached on Monday through Friday 8:00 AM to 5:00 PM EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Daniel Ryman can be reached on (571) 272-3152. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

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/Christopher Crutchfield/
Examiner, Art Unit 2466
9/8/2010

/Daniel J. Ryman/
Supervisory Patent Examiner, Art Unit 2466